Flexible media transport framework based on service composition for Future Network

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1. Introduction

The International Organization for Standardization (ISO) is performing tight work on Future Network. Concretely, in the workgroup 7 of the JTC1/SC6 Technical Committee. Future Network is defined as the network infrastructure in which communications are going to be sustained in the next 15 or 20 years.

In this regard, nowadays there are two big major trends to face the Future Network design, two approaches that at a first glance may be at odds. The former is known as evolutionary and the latter is named revolutionary or clean slate. It must be said that they are not exclusive one to each other. Clean-slate approaches bring novel solutions to solve the shortcomings of current Internet. These new fresh ideas must feed the evolutionary research in order to blend both leads to the next future network. A non-backward compatible architecture, or non able to provide an interface with the current Internet is likely destined to fail, because stakeholders play an important role in the deployment of new infrastructures, now and in the future, and they are not just willing to throw away their current investments. Thus, any new architecture must consider the current Internet as an starting point and not as a old-fashioned and useless network.

However, Internet is evolving towards a content exchanger, a place for creating and consuming experiences over time, either live or offline. People, other important actor, demand live multimedia communications, new experiences and better quality, anywhere at anytime with any kind of interface (network and/or device). So, Future Network will be oriented to user media experiences. Requirements for Future Network include mobility, multi-homing, ubiquity, trustiness, security, robustness, context/content-awareness, increasing of multimedia experiences, and evolvable.

Future Internet must be based on the premises of simplicity, flexibility, evolvability, backward-compatibility with legacy technologies (or it must be provided a well-know interface between both), and focused on high quality multimedia communications, as the basic traffic and lead for the future networking. This work introduces common guidelines defined in several standardization organisms towards future networks based on the actual mechanisms and protocols used to treat the multimedia data, most of them placed in the application layer of the OSI reference model. To face this challenge service-oriented architectures offer a flexible approach, which enables to define services and compose them both, in run-time or design-time, to fit the requirements for particular media communications over heterogeneous context, for any kind of media content, either time-dependent or time-independent. Future Network will go further than the application layer and go down to the communication protocols themselves, choosing in a dynamic fashion which kind of basic services (e.g.: acknowledgement, sequence number, flow identification, congestion windows, etc) and media services (e.g.: content adaptation, scalability, transcoding, etc) are needed in a particular communication, according to the parties capabilities and the media to be transmitted requirements. Thus, this work introduces the flexible media transport framework, as the media service composer element, forming a common container with just the metadata needed, to compose and dynamically adapt specific media services for a given communication for every sort of content.

2. Basics Concepts Definition

2.1. Data

Meaningful bunch of bits which either are the basic blocks to create content or are simple raw information units to be handled.

2.2. Content

Content is defined as meaningful media data that is carried in the payload of datagrams sent over the network. Content is media and is classified into two categories, time-independent media object and time-dependent media objects. The former are those that the semantic of the content does not depend upon a presentation according to the time domain (ie: text, image). The latter are those that exist a temporal relation amongst the media units (ie: multimedia video stream). Continuous media is a kind of time-dependent media object when media units are presented consecutively with the same temporal gap between consecutive units. Regarding to traversing the network, content may be classified as:

(a) "static" content (time-independent), with neither network time dimension (asynchronous transmission), such as text and still images (photographs)

(b) "streamed" or "offline" time-dependent content which has a time dimension when rendered but does not have requirements for latency across the network, such as MP3 files or Video-on-Demand, and "live" time-dependent content which has latency requirements across the network, such as in telephony and video-conferencing.

2.3. Container

It is defined as the encapsulation structure for either data or content. Container is composed by a payload and one unique header which has two differentiated parts, one regarding to the application data and another one more often variable along the route regarding to the underlying network. Container has attributes as header fields, which some are related to particular services, and others are general and specific for a sort of communication.

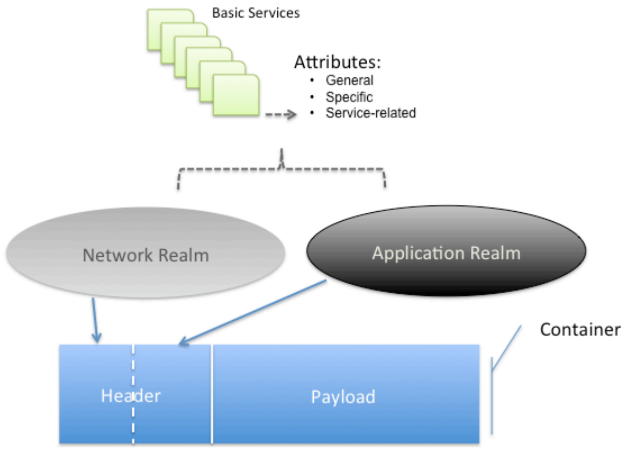

Figure 1: Container

2.4. Content adaptation and scalability

Scalability and adaptation enable to adapt any kind of content to a particular context. Content adaptation means to modify in some kind a given input in order to generate an adapted output according to a particular context, such as the capabilities of the other partner in a videoconference call. Adaptation can be performed in the spatial, temporal, quality domain, as well as simply modifying the codec applied in the case of audiovisual content.

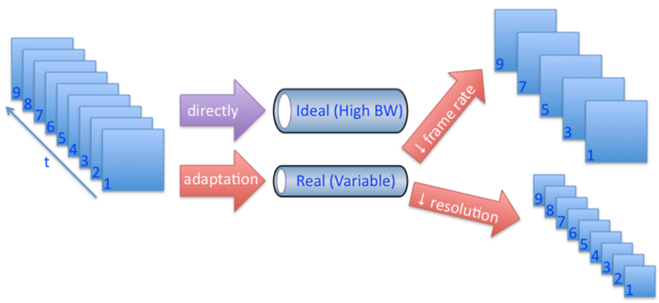

Figure 2: Media Content adaptation scenario (particular case of video)

The term scalability is defined as the ability of coding a video bit stream with different qualities or spatial / temporal resolutions of the same original content, in such a way that some structured parts, named subsets, can be removed from the bit stream giving as a result the same content in lower fidelity. Typical scalability modes are spatial, temporal or quality.

Layered Coding (LC) is based on the idea of splitting a video stream into several hierarchical layers consisting of a base layer, with minimal features, and enhancement layers. The more layers received the more quality achieved, as a receiver-driven adaptation system. The main drawbacks of this coding are basically two: (1) real implementations of LC, such as wavelet-based solutions (DWT, e.g.: JPEG2000 and Dirac Pro) and SVC (Scalable Video Coding), require enormous computational complexity and entail high delay, and (2) its weakness in terms of reliability in error prone environments.

Instead, Multiple Description Coding (MDC) provides error resilience to media streams by dividing the source into, at least two, selfcontet subsets, referred as descriptions, which are issued by different paths over the network. Depending on the number of descriptions received a lower or higher quality of the content is achieved. The robustness of the system lies on the assumption that error bursts are more common than singular ones, in such a way that, in the case of images, errors in different descriptions are scattered along the whole canvas in the reception side. There are several proposals of MDC based on balanced (same importance) and unbalanced (different importance) descriptions, but few suitable to match the strict requirements for HD conversational real-time services. The main factor to choose the best MDC technique for covering critical constraints of delay and scalability is basically the simplicity, which entails low processing complexity, hence low delay for these conversational services.

2.5. Context-Awareness

The Future Networks shall be aware of context. Three important aspects of context are: where you are; with whom you are; and what resources you are nearby. For example, context-awareness is applied to mobility, it refers to a general class of mobile systems that can sense their physical environment, e.g., their context of use, and adapt their behaviour accordingly. Context awareness is applied to network entities that are aware of any information (e.g. context) that can be used to sense and react based on the environment. The context includes but not limited to the user, device, service, system resources, kind of content to be transmitted and network context. The user context can include user characteristics, user's location, user's preference, and environmental constraint of user (e.g. public are where silence is required, working place, home, etc.). The device context can include type and capability of the device. The service context can include service availability, required QoS level, and service performance. The system resource context can include CPU, memory, processor, disk, I/O devices, and storage. The kind of content context can include the possibility to buffer information (delay tolerant), minimum QoS/QoE required, content adaptation preferred, necessity of integrity and/or confidentiality. The network context can include bandwidth, traffic, topology, and network performance. The Future Networks should support the context management to provide customized and context based services.

2.6. Layered vs. modular model

The layered nature of the TCP/IP stack is discussed, advocating greater flexibility in network architecture. Thus, most of the different proposals are of disruptive nature, arguing for "clean slate" approaches where protocol design is usually based on micro-modularizing networking protocols into its fundamental (atomic) functions. These functions can be combined according to the requirements of a communication, avoiding particular solutions that add complexity to the architecture.

Services are well-defined and self-contained functions, used to establish and manage communications. The idea is to create new protocols using these atomic functions or services as basic building blocks. This trend is closely related to Service-Oriented Architecture computing paradigm. The goal is to compose services into a workflow for obtaining a more complex service. This way, our proposal uses ensembles of basic services to provide advanced communication services.

2.7. Content-Aware Based Congestion Control

Common congestion controls are focused on adapting the data rate at bit level according to the network available resources, instead of on adapting the content (semantic approach). For example, windows-based congestion control is based on a transmission window, such as TCP that limits the amount of data to transmit following usually an AIMD (Additive Increase / Multiplicative Decrease) algorithm according to the network status. Other congestion control approaches closer to the multimedia world, such as rate-based, media-aware rate control and receiver-based mechanism, act over the sending data rate in a smoother way than the strict data based congestion controls, but they also treat the video flow as a bit stream adapted to the network conditions. On the other hand, the CBCC is focused on the experience of the user, applying a different control according to the media content conveyed; so, what does the user want to receive, the media content. A bit-stream oriented congestion control is used for time-independent media where a reliable and fast communication is pursued. On the other hand, for time-dependent media the adaptation is done basically over semantic content parameters, instead of over the bit stream. Both strategies act over the output data rate, increasing or decreasing the amount of bits issued according to the network congestion, but taking care of what sort of content is conveyed. For example, if the content is a video stream (time-dependent media), according to the variable conditions of the network and taking into account the nature of the data transported, the output data rate adaptation is directly related to the adaptation in real time of the video source parameters, such as either the resolution, the frame rate or the codec, instead of directly decreasing the data rate.

2.8. Error resilience

Currently, most error resilience techniques applied on real-time communications are FEC (Forward Error Correction), which adds redundant information increasing the data rate. The application of interleaving mechanisms increases the probability of recovering lost multimedia data in the presence of bursty losses across the network. These techniques can be applied according the data conveyed as a modular services in an atomic architecture.

2.9. MANE (Media Aware Network Element)

It is a content and context aware network element able to treat the media content passing through to accommodate the content and service related according to the context. It is forecasted as one of the key elements in the edge of networks. This element may handle all attributes of containers, both application-related (likewise current transport and application layer) and network-related (likewise current network, data link and physical layers).

2.10. QoE / QoS

QoS is usually defined regarding to three parameters: bandwidth, delay and error. In conversational communications, bandwidth consumption is related to the chosen technology (although the lower one is desired), the delay has to be bounded to assure conversational interactivity among participants, and should be provided a high error resilience to assure a good data delivery in front of any change on the network.

Instead, QoE is related to how the user perceives the received content.. QoE is more related to subjective quality estimation rather than objective measurements, although they can be related by different mapping schemes.

2.11. Connection-oriented and connection-less

Any communication, whether artificial or biological, can be performed following two different schemes or modes. The fact is that the aim of communication is just to transmit something from one point to another, thus the way in what this occurs can vary over time. Basically there are two modes, (1) connection-oriented, based on a connection process where participants are aware of the presence of each other before transmitting the useful information, instead of (2) connection-less where data is straightforward issued without any kind of previous establishment. Example for connection-oriented is a telephone call, and for connection-less a radio broadcasting.

2.12. Service Composition

Service composition is a cornerstone process involved when providing services. It can be said that service provisioning is divided in a) Service Definition, b) Service Publication, c) Service Discovery, Service Composition and d) Service Adaptation. Concretely service composition consists on the process of selecting, allocating and combining those services, to be executed along the path to the service provider node. Selection and allocation decision is done taking into account the cost of using them, with regard to the requester priorities. Those required ASs are distributed along the end-to-end path nodes trying to avoid unnecessary duplicities and thus, redundancies. Once atomic services are composed, they are orchestrated in the form of workflows, thus, obtaining more complex services.

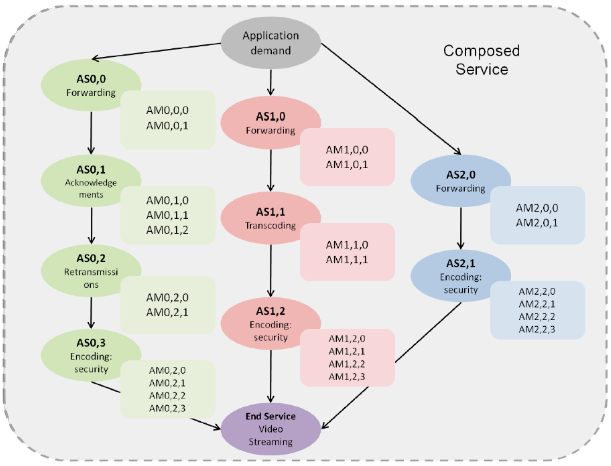

Figure 3: Service Composition procedure

Figure 2 shows different possibilities to obtain a Video Streaming service based on different configurations of Atomic Services (AS). It is also possible to see that each AS can have several independent implementations, called Atomic Mechanisms (AM). The concatenation of ASs leads to a Workflow (WF).

Regarding to the dynamism of the composition, note that it is possible to identify two types of composition: (1) Static, where services to be composed are selected at design time and (2) Dynamic, where services to be composed are selected at runtime. The latter type is much more challenging than the former, even more, if it's required in a fully-distributed and heterogeneous environment.

Once ASs are allocated along the end-to-end path, the service communication is set up. Communication conditions may vary, thus, the adaptation process is required to be dynamically executed at runtime to react to context changes. This may imply a change on WFs in order to improve service performance, overall Quality of Experience (QoE), Quality of Service (QoS) and resource usage. Several approaches can be adopted for service composition. It would be interesting to propose benchmarks and comparisons of composition algorithms and techniques, in order to determine which are the best under specific conditions.

3. Internet traffic trend

Few years ago, after the revolution of the WWW and the spreading of the Internet to the end user, the highest percentage of the traffic in the backbone was HTTP. Recently, this tendency has changed towards the P2P traffic and the exchange of media data between users or providers and users. This change of the habits of users affects to the concept of the usage of the network (nowadays process is in the ends not in the core of the network), the model used to communicate the different users (server – client model vs. P2P model), and the type of contents transmitted. Actually, current Internet is already a media network based on P2P traffic and on video on demand or videostreamng applications, where a little part of the Internet users generates the major part of the global IP traffic. In 2009, media content, from P2P and VoD services, represented the 80% and 90% of the global Internet traffic respectively. The convergence of television, video, graphics, audio and networks is a fact and it is expected that video traffic will increase even more and more, for instance, thanks to the increasing demand of HD or 3D media. The limitation is on the current state of the Internet and the associated hybrid-networking infrastructure that requires an enormous amount of effort and fine-tuning to transmit high quality media.

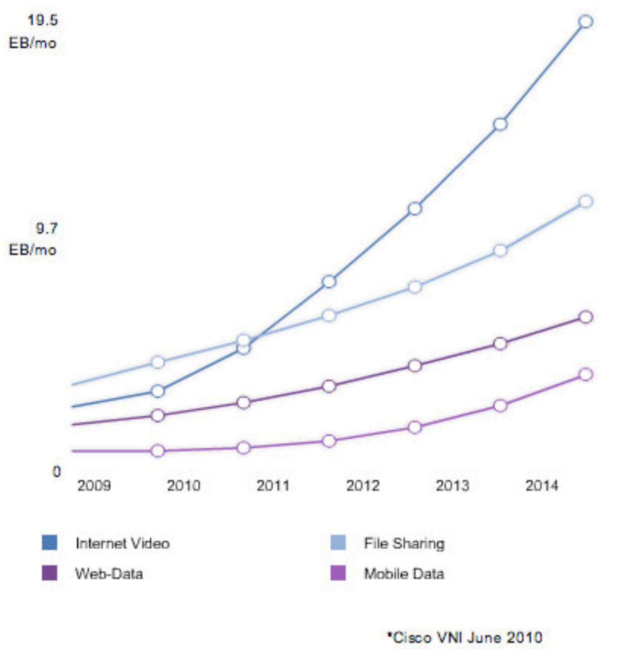

Figure 4: IP network forecast according to CISCO Visual Networking Index

It must be remembered that current network started as a computer networking revolution, and now the Internet is evolving as a user driven network more and more, based on audio-visual media content of all kinds. So, the change in our traditional paradigm of an information and communication network which relies on computers and telecommunications is clear towards a new media network, for sharing all kind of cultural knowledge, science, technology, arts and games.

Users are becoming into active players thanks to media networks. An evidence of this is the growth of the contents generated by the users.. Their interest in media content is growing amazingly, but will do it more and more in the next future. That means the people are the new driving force in the design of future Internet and introduce new requirements and demand for new services, applications and functionalities.

So, the Future of Internet must be thought starting from this new reality, where users are continuously consuming media (music, TV series, etc.), both time-dependent and time-independent, and beginning to offer self-generated videos. In fact, people at home will create soon high quality content, where the user can interact with the content and, even, generate and modify it in a collaborative manner. The problem is that the current Internet is not ready for this convergence between the real media world and networks and future demands.

4. Current Internet

The development of networks during the last years has shown that it becomes harder to integrate new functionalities in order to fulfill the demands of new applications and the capabilities of new transport technologies. Especially the core mechanisms are hard to change as it lies in a rigid and ossified architecture.

The current picture of the networks shows a large, heterogeneous, dynamic and complex distributed system. Lots of patches aimed to amend different issues that have arisen during last years. Current networks has to deal with new services, applications and computing paradigms such as new modes of interaction, identification, context-awareness, energy efficiency, seamless service discovery and composition, mobility, ubiquity, etc.

Current multimedia research activity has been and still is focused on how to adapt the media to the running network architecture, usually defining some mid layer that provides particular features for this sort of traffic. RTP/RTCP, RTP for uncompressed video, MPEG-TS, and so on, are well-known examples of these mid layers designed for this reason, to adapt the media content to the TCP/IP network. New mid layers are in continuous evolution to adapt themselves basically to users and underlying network requirements. All these mid layers are designed to adapt the media data to the Internet, over the classical stack of communications designed in the 70s to transmit computer data from end-to-end user.

The mid layers or protocols are presented in the graphic below. From the left side, the OSI model, the TCP/IP stack and the protocols adopted for the media transport.

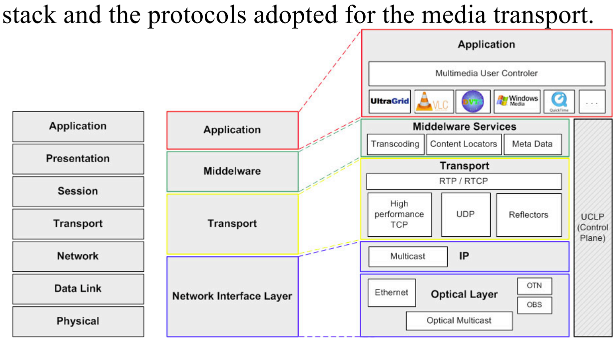

Figure 5: Current mid layers or protocols used for media

First off, TCP/IP architecture has two main features that any future network should accommodate: simplicity and flexibility. TCP/IP also just takes care about the end-to-end communication, how to identify connections globally and the path among them as well as to control the application bit-stream, without having any strict communication with lower or higher layers. So, TCP/IP stack is independent enough to perform its functions over any sort of network (Ethernet, ATM, X.25, SDH, etc.) and to allow any application to use its global services. These two simple points have made the TCP/IP network to prevail over the others as the main network of the networks. Nevertheless, a big drawback of TCP/IP, is the routing protocols, because of mainly its stiff operation and maintenance of the routing information in the network nodes. Nevertheless, a big drawback, as the sort of "control plane" of TCP/IP, is the routing protocols, because of mainly its stiff operation and maintenance of the routing information in the network nodes.

Moreover, Internet is evolving towards a media content exchanger, either time-independent or time-dependent, a place for creating and consuming data over time, either live or offline. People demand live multimedia communications, new experiences and better quality, anywhere at anytime with any kind of interface (network and device). Engineers must propose extensible and scalable solutions to allow the adoption of new requirements and functionalities not foreseen yet pushed by users and service providers.

Current network architecture design is based on a hierarchical layered model like OSI or TCP/IP stacks. In these models, networking functions and protocols are grouped in layers, according to a common objective and scope. Thus, each layer performs different networking tasks, restricting inter-layer communication to immediately adjacent layers. In theory, each layer is in charge of a group of functions, but in practice functions overlap at different layers, adding protocol overhead and blurring the layered structure of the protocol stacks.

To send an RTP packet it is needed 20 bytes of the IPv4 header, 8 bytes of UDP and 20 bytes more of RTP. So, 48 bytes assuming that there is not additional information (optional fields), many of them useless today. For IPv6 (40 bytes) the RTP headers add up to 68 bytes, and a TCP/IPv6 acknowledgement packet is 60 bytes, though it only carries about 4 bytes of useful information. It must be remembered that TCP/IP network was designed to work over any sort of network, and for any sort of application. The main operation of TCP/IP is to transfer data from end to end, by two different ways, connection oriented (TCP) or connectionless (UDP).

As example of useless fields on the current stack, when MPEG-TS (well self-structured media format with several information used to "transport" media stream) is sent over the network, a lot of headers and additional information about the MPEG stream are added. Some fields are already present in the RTP header, increasing the amount of information, resulting in a waste of time and resources. This is an important hint that these protocols need a hard redesign or to be replaced. Historically, new features have been added by means of inserting extra information in higher layers, instead of squeezing the capacities and features of the existing ones.
Now, there are heterogeneous scenarios, either devices or interfaces, applications and network technologies, which require new modes of interaction amongst peers (nodes, applications, services…) of the network not covered by current network architecture. So, network services should evolve on an architectural framework to become flexible, ubiquitous, composable, dependable, secure, context and content aware, and adaptable in execution and design time.

The architecture presented in this document is based on the premises of simplicity and flexibility (evolvable) according to the current Internet, compatible with legacy technologies, and focused on high quality multimedia communications as the basic traffic and lead for the future networking.

5. Media Transport based on Service Composition

The model is designed as a service-oriented approach for a flow-oriented context-aware network architecture working mainly in a connection-oriented mode, although connection-less is allowed for particular sort of services, where communications are composed in situ (using reusable components) according to the needs and requirements of the consumed service.

The architecture is able to work in connection-oriented and connection-less fashion with datagram transmission, depending on the working environment. In current Internet, the most data of Internet is conveyed by using TCP with its strict flow and congestion control. Even media applications using UDP transport protocol relay either on RTP/RTCP, as transport / control protocols, or on a particular control scheme at application level. So, future network is designed to work with both modes, mainly as a connection-oriented but also as a connectionless mode for particular sort of communications, which do not care about if there is any peer on the other side, such as short messages services (SMS and tweets). Broadcasting services may be set as connection-oriented communications following a multicast pattern across the network, or connection-less in simplex communications such as traditional radio / TV.
Services are classified into basic (atomic), and composed services. Basic (atomic) services are those individual functions commonly used in networking protocols (e.g. acknowledgments, sequence numbers, flow control, etc). These are well-defined and self-contained functions, used to deliver data in a self-adaptable, self-configurable and context-aware way. Concretely, media services are those atomic ones which operate with multimedia mechanisms (such as transcoding, CBCC, protection, etc.) that belong to the content realm and that they may be executed by the same peer or by another able to perform the task in order to provide a higher level media service. Composed services are the result of combining basic services. Each composed service or application implies consuming different basic and, sometimes, other composed services; appearing possible dependences between them. Also, they can involve one or more nodes, depending on the complexity of the service. As result it is generated a container for each particular communication.

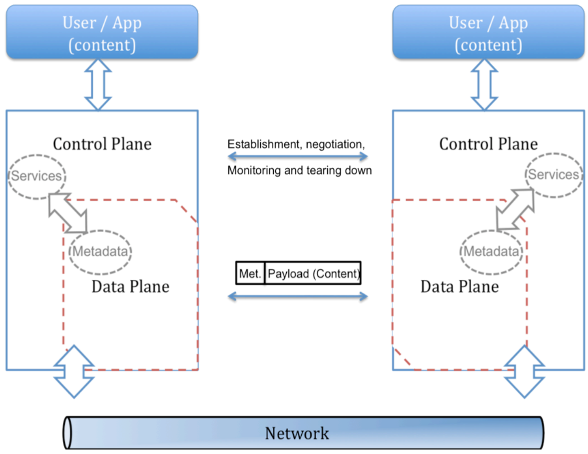

Figure 6: Media Transport FN architecture

The architecture consists of two planes, the control and data plane. Control plane is in charge of the establishment, negotiation, follow-up, and the tearing-down of a flow connection and enables management tasks. Once a service is composed, it results in a PDU formed by a payload plus the metadata referred to each particular basic service in the composition. Data plane is in charge of the transmission of the PDU generated from one peer to another following de connection.

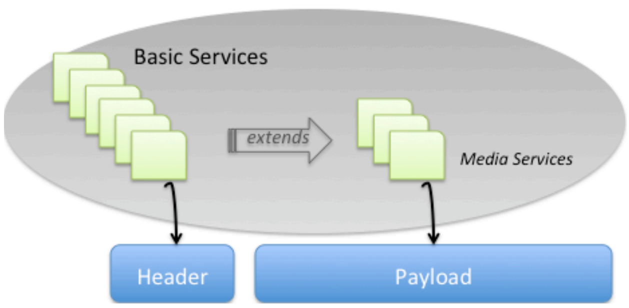

Figure 7: Media Tansport Service types

Some related atomic services can be grouped and labelled, in such a way that just one identification field represents the use of all the basic services in the group. Thus, negotiation can be done either one by one service, or by service group. For example, for a media transmission, there are some basic services that may be grouped such as sequencing, timestamp, acknowledgement, encoding, etc… under the label “media_transmission”. There is a direct relation between the communication mode (connection-oriented and connection-less) and the basic services to compose, in such a way that in some services used in a connectionless communication, can be sign aled in the control plane in the connection-oriented mode, thereby saving space in the packet (or comple xity in the end systems)

6. Use Cases

6.1. HD Multiparty videoconference

Multiparty videoconferencing involves several participants who may have as wide variety of capabilities as basically heterogeneous networks and devices. One possible proposal to support multiple parties with different capabilities (different context) is enabling scalability to adapt the video to the heterogeneous environment, by means of, for example, a Polyphase Downsampling Multiple Description Coding technique that is applied on the video content to generate several lower resolution balanced subsets from the original video source, which will be issued over the network into self-contained separated flows. Thus, with this solution a wide range of devices can be covered with the same video source, in such a way that the more subsets delivered the more fidelity is achieved.

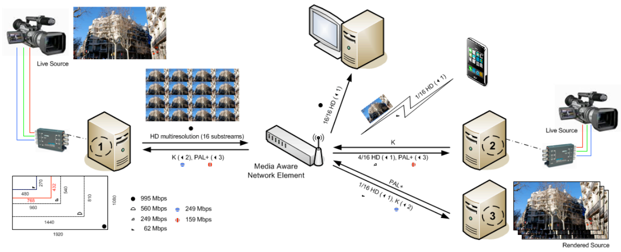

Figure 8: Multiparty HD scenario

Currently, such scenario is composed by two planes, a control plane and a data plane. Data plane works over a RTP/UDP/IP stack with either some sort of extra signalling to control the synchronization, or an extension to the RTP header to differentiate and synchronize the multi-streaming generated for this technique.

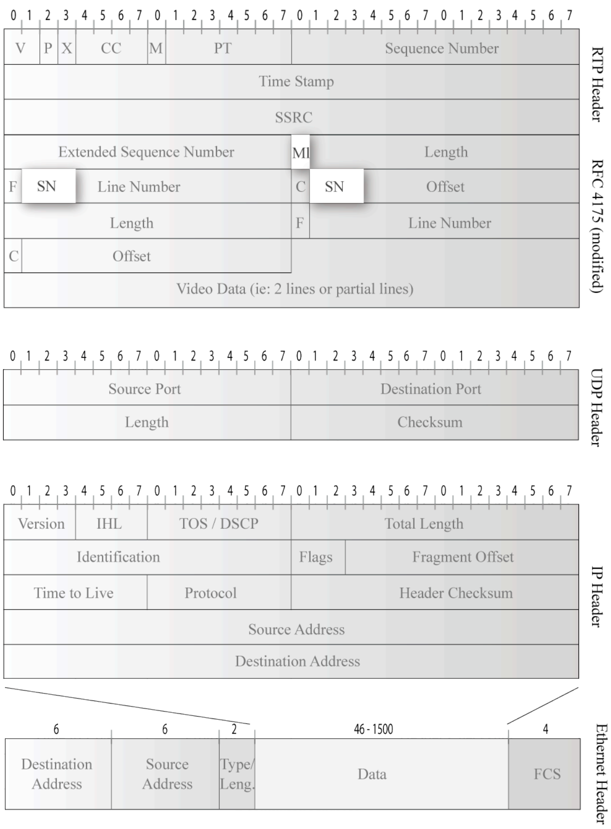

Figure 9: Protocols stack for multistreaming communication

Future Network will be a content aware network able to provide a composed service to fit the specific requirements for a given communication. Firstly, a control plane is also needed to establish, negotiate, monitor and tear-down the communicaton among the involved parties.,Then, a data plane to transport the data is required. During the negotiation phase, current multimedia signalling application protocols determine the capabilities of the parties in the call, entailing which kind of media will be issued over the network infrastructure. Future Network should go further than the application layer and go down to the communication protocols themselves, choosing in a dynamic fashion which kind of basic services (ie: acknowledgement, sequence number, flow identification, congestion windows, etc) are needed in a particular communication, according to the parties capabilities and the media to be transmitted requirements, and then work just with those services needed In such a way, for the scenario presented, and taking into account the current communication stack for this kind of traffic as a reference, basic services that could be negotiated amongst the parties can be:, sequencing, sub-stream synchronization, content handling, timestamping or global time referencing QoS labelling, FCS of all data, content based congestion control, and back reporting with media statistics. These services entail related attributes as part of the header fields of the container. Besides, there will be also general attributes for any communication, such as media type (content to which a particular composed service is provided), parties identification, data length. And specific ones associated to the kind of transmission such as line number, data pointer (with regard to the content information conveyed, current offset) and media unit reference (current M bit to signal the end of a media unit of the media object, in this case a frame), That results in a bunch of attributes, which conform the header of the total container, to support each of these basic services. Using this methodology replication of functions is avoided and just used those functions that are needed to support a particular communication in a flexible way.

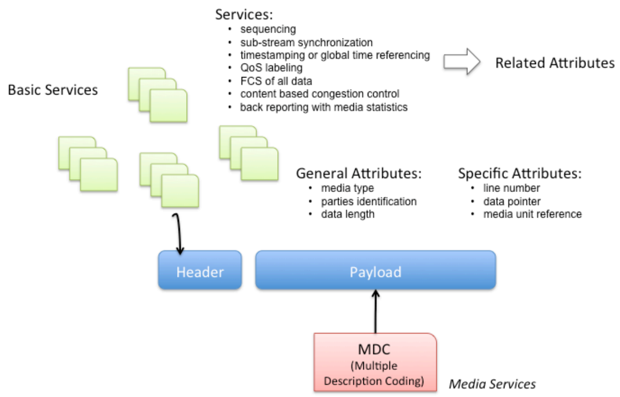

Figure 10: Service for multistreaming communication

6.2. Web browsing

Multimedia contents consume the largest part of the current traffic carried in the network, mainly because of p2p applications of file sharing, live streaming and lately the Video on Demand application. In this scenario, the second majority traffic on the network is reserved to web browsing, an asynchronous reliable service of data transfer between two nodes in the network following a client/server model. Actually, this is the kind of service for which the current TCP/IP network was primarily designed, to provide a reliable (using TCP) or fast (UDP) communication between peers working over non-reliable physical networks.

Currently web browsing uses HTTP/TCP/IP, which performs a strict data control in order to assure the complete data transfer between the server and the client with no errors (integrity). This reliable communication is achieved by means of a connection-oriented protocol at transport level, the TCP, which performs also a strict control, in terms of flow and congestion.

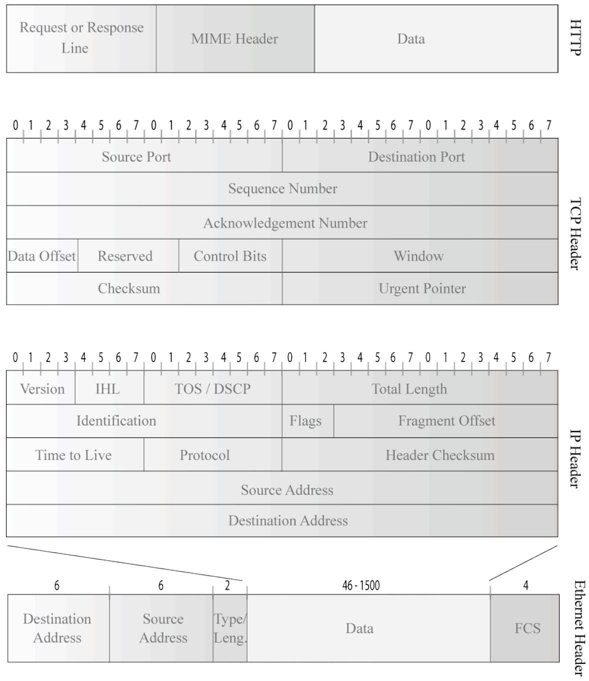

Figure 11: Protocols stack for web browsing communication

Future Network will be a content aware network, by means of the control plane, which in the negotiation and establishment process shall specify the sort of media content (dependent or independent). Hence, in this scenario the basic services will be those that identify the peers and assure an asynchronous reliable data transfer. Basic service shall be:, sequencing, content handling, QoS labelling, FCS of all data, content based congestion control, and acknowledgement. General attributes shall be also media type (content to which a particular composed service is provided), parties identification, data length. Specific attributes, regardless the basic services' and general ones shall be those related to current HTTP attributes.

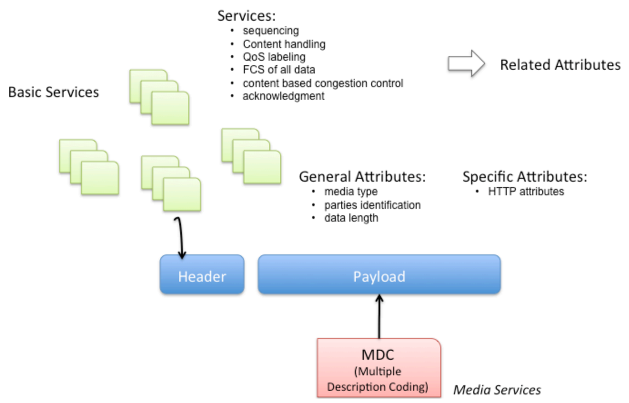

Figure 12: Service for web browsing communication

6.3. Media Aware Network Element

6.3.1. Content based congestion control

Media Aware Network Elements are intended to be the network elements in charge of higher functions related to media instead of simple data relays. MANE are able to be aware of the content conveyed through the streams crossing it and then reacts over them according some rules and depending on the sort of media content itself in front of network event, such as congestion.

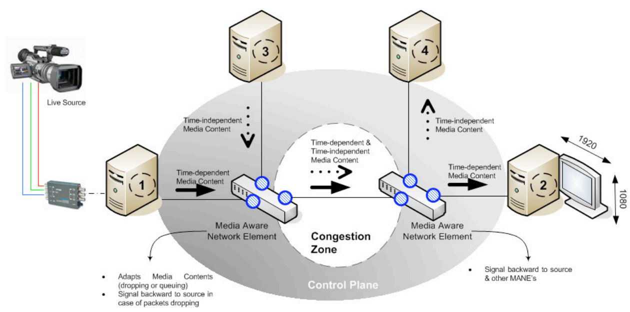

Figure 13: Media Aware Network Elements reacting in front of congestion

In the scenario presented in the above picture there are two main streams, one conveying time-dependent media data and the other time-independent. Thus, in case of congestion in the network MANE's reacts adapting media contents. In the case of time-independent content data may be queued and sent at bursts instead of following a continuous stream. In turn, for time-dependent media may be performed different actions depending on the capabilities of the MANE and the content itself, such as either dropping particular packets of an scalable content or adapting the content to the network status. In both cases, a signal of congestion is sent backward to notify the source.

6.3.2. Decision-making

Nowadays heterogeneity is the key concept to define the network, and scalability and adaptability solutions to reach users behind different devices and interfaces. MANE should be the network element in charge of making the decision, according to the control plane. For example in the case of video transmission, will be in charge of adapting the content or relaying as many sub-streams (from a scalable video coded video stream) as needed to each peer behind it. In the scenario below is presented a multistreaming transmission of a video content using a MDC scheme to split the content into several pieces in order to reach as many users as possible. MANE is the element which relays the particular pieces to each peer according to their capabilities and status reported by the control plane.

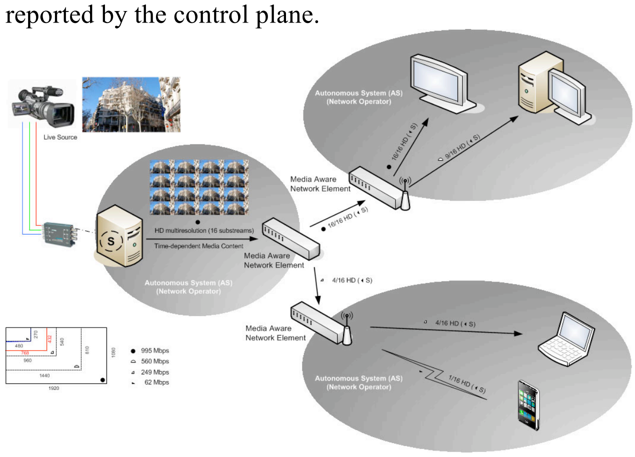

Figure 14: Media Aware Network Elements relaying just media data needed

6.3.3. Seamless mobility

A user may be watching an online TV program on his/her smartphone connected through a cellular network operated by Operator A, when goes into his/her home and wants to handoff the radio stream to the TV, which is connected to the LAN operated by a cable Operator B. Media mobility inside the same LAN can be performed be the MANE placed inside the LAN, operated by the same manager. In the case of different operators, MANE has to be placed in the interconnection between their own networks and perform the handoff in that point. This is an important point today, because user does not care who provides the service, he/she just wants to consume a specific media content seamless everywhere and with any device.

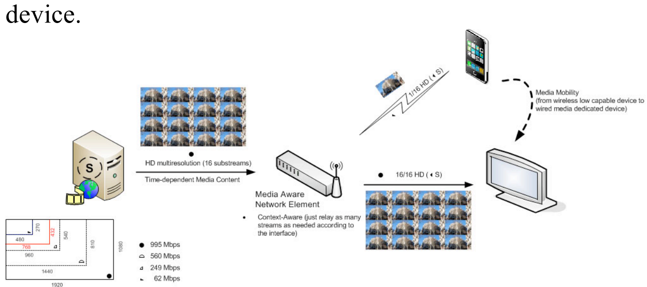

Figure 15: Media Aware Network Element performing media mobility inside an AS

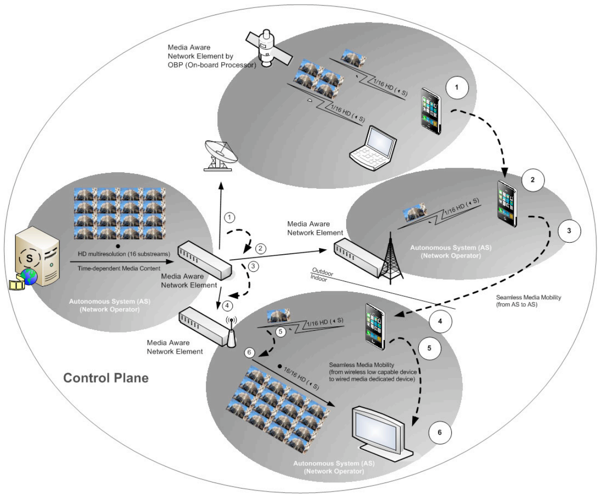

Figure 16: Media Aware Network Elements performing media mobility amongst several AS's

7. Conclusion

Current Internet is based on a rigid layered model which makes difficult to add new functionalities introduced by the appearance and evolution of services and applications. During its 40 years of life, some limitations have been observed when new requirements in communications appear. Requirements such as security, mobility, multihoming or multicast were not foreseen when current TCP/IP stack was born. This fact motivated the evolution of Internet by means of patches which break the end-to-end principle followed in its inception. Moreover, Internet traffic is changing pushed by new applications. In its origin, communications were designed for carrying the information generated by email or web pages. However, nowadays, media applications generate the major part of the Internet traffic. These applications introduce new challenging requirements difficult to provide by current networks. Not only increasing the required bandwidth, they impose stringent timing constraints for example in the case of real-time and interactive video communications. It is foreseen that Internet will become a powerful platform to deliver media contents anywhere and anytime. The main problem here is how to provide media contents to users according to their specific needs. In this sense, current Internet makes this a difficult task. Current solutions face specific problems and introduce complexity and inefficiencies to current model. Thus, some proposals are arousing to fulfill this situation from scratch. These solutions, known as clean slate, propose new architectures to overcome current Internet limitations. A promising approach avoiding the rigidity of layers is to introduce service-oriented architectures, which by means of service-composition is able to provide customized, flexible and evolvable network architecture, allowing either to be backward compatible or to have a well-know interface between both networks, as needed to facilitate the transition between current and future network more straightforward for both communities, people and stakeholders. These kinds of architectures are grounded in the use of services which can be combined into more complex ones according to communication and user needs. Some projects regarding these issues are 4WARD, RNA, SILOS and FIND.

Currently, there are several international organisms such as ISO/IEC (JTC1/SC6), ITU-T (SG 13), IEEE (NGSON WG) which are putting efforts to propose standardized architectures to overcome the current Internet model and to pave the ground to its evolution to meet requirements, services and applications, bearing in mind all the lessons learnt from Internet TCP/IP model.

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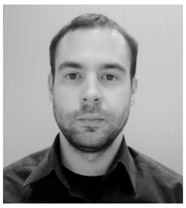

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